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IN REPLY REFER TO: Joint Interoperability Test Command (JTE) **12 Nov 15**

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Extension of the Joint Interoperability Certification of the Cisco Enterprise Session Controller (ESC) 8

References: (a) Department of Defense Instruction 8100.04, "DoD Unified Capabilities (UC)," 9 December 2010
(b) Office of the Department of Defense Chief Information Officer, "Department of Defense Unified Capabilities Requirements 2013, Errata 1," 1 July 2013
(c) through (e), see Enclosure

Certification Authority. Reference (a) establishes the Joint Interoperability Test Command (JITC) as the Joint Interoperability Certification Authority for UC products.

Conditions of Certification. The Cisco ESC 8; hereinafter referred to as the System Under Test (SUT), meets the critical requirements of the Unified Capabilities Requirements (UCR), Reference (b), and is certified for joint use as an ESC in Type 1, 2, and 3 environments and as a Local Session Controller (LSC) with the conditions described in Table 1. This certification expires upon changes that affect interoperability, but no later than 27 June 2017, which is three years from the date of the original UC Approved Products List (APL) memorandum. Desktop Review (DTR) 23 was requested to include the 8845 and 8865 Voice and Video over Internet Protocol (VVoIP) phones. See "Test Details" paragraph for more details.

Table 1. Conditions

Condition	Operational Impact	Remarks
UCR Waivers		
None.		
Conditions of Fielding		
None.		
Open Test Discrepancies		
The SUT video end instruments include H.323 proprietary ROUTINE only end instruments depicted in Table 4. Additionally the SUT includes a Jabber client that offers video and voice; however, during the original test the Jabber client was certified for audio only as a soft phone and for XMPP IM&P as an XMPP client.	None	CLOSED See note 1.
Per the vendor's LoC, the SUT does not display weighted Terminal Coupling Loss (TCLw) and equipment impairment factor in their call detail record (CDR).	Minor	See note 2.
Per the vendor's LoC, the does not fully meet separate video and voice ASAC counts.	Minor	See note 2.

Table 1. Conditions (continued)

Condition	Operational Impact	Remarks
Open Test Discrepancies (continued)		
The SUT does not properly handle signaling events when setting up an inter-switch V.150 secure call with Avaya Communication Manager (CM) 6.0.	Minor	See note 3.
Per the vendor's LoC, the SUT proprietary video EI does not provide the ability to enable or disable the transmission destination unreachable msg.	Minor	See note 2.
Per the vendor's LoC, the SUT fails to immediately divert all precedence above routine calls placed to ROEIs. The SUT diverts only when the ROEI is busy if it is idle it will offer the call and divert if not answered.	Minor	See note 2.
During the original test, video calls between SUT H.323 PEIs (C90/EX90/SX20) and other UC video endpoints dropped at approximately 30 minutes.	None	CLOSED See note 4.
The SUT fails to answer with correct payload number per RFC 3264. Instead, of responding to the V.150.1 payload numbers in an SDP, offer the SUT always responds with payload number of 118 and 120 for State Signaling Events (SSE) and Simple Packet Relay Transport (SPRT) respectively which prevents successful secure call attempts.	Minor	See note 3.
Per the vendor's LoC, the SUT does not support an AS-SIP ESC to EI signaling interface.	Minor	See note 3.
Per the vendor’s LoC, the SUT supports Primary Rate Interface requirement to be in compliance with ANSI T1.619-1992 and T1.619a-1994 with following exception, NFAS is not supported.	Minor	See note 2.
Per the vendor’s LoC, the SUT does not support Public Key Infrastructure Requirement IA-049030.	Minor	See note 5.
Per the vendor’s LoC, the SUT does not support Confidentiality requirement IA-069040.	Minor	See note 5.
During the original test, the SUT 9951/9971 voice/video SIP ROEIs did not fully support inter-enclave hold feature while video enabled.	None	CLOSED See note 6.
Per the vendor’s LoC, the SUT does not support a persistent TLS connection between AEIs and the enclave fronted SBC because the SUT does not support AEIs.	Minor	See note 3.
Per the vendor’s LoC, the SUT video conferencing system does not support all required audio codecs. The SUT does not support the G.723.1 audio codec.	Minor	See note 2.
Per the vendor’s LoC, the SUT partially complies with the EDS gateway requirements per SCM-005300.	Minor	CLOSED See note 7.
When the SUT MCU 5320 places an outbound video call to other SUT C90 and SX20 video endpoints in either environment 1 or environment 2, the call drops at exactly 15 minutes.	Minor	See note 3.
The SUT SX20 and C90 configured on environment 2 are not able to establish two-way video calls with the Polycom RMX UCCS. The SUT SX20 configured on environment 1 or environment 2 is not able to establish two-way video calls with the Vidyo UCCS. These anomalies occur when the SX20 and C90 are registered to the ESC Environments and do not occur when these endpoints are registered to the LSC.	Minor	See note 3.
Per the vendor’s LoC, the SUT does not correctly respond to stream errors. Instead of responding with a stream error and closing the stream, the server terminates the connection non-gracefully.	Minor	See note 3.
Per the vendor’s LoC, the SUT does not generate a new Client-to-Server Stream. Server reuses the old stream ID instead of generating a new stream ID.	Minor	See note 3.
Per the vendor’s LoC, the SUT does not include empty element in its advertisement of the SASL.	Minor	See note 8.
Per the vendor’s LoC, the SUT does not fully comply with SASL failure requirements. The SUT does not comply with requirements IM-000710, IM-000720, and IM-000730.	Minor	See note 3.
Per the vendor’s LoC, the SUT does not fully meet deleting a roster item requirement. The SUT does not comply with requirements IM-001310 and IM-001320.	Minor	See note 3.
Per the vendor’s LoC, the SUT partially complies with rules for Server Processing of Outbound Subscription Requests. The SUT does not comply with requirement IM-001350. Server sends presence type “unsubscribed” with status Not Found.	Minor	See note 3.
Per the vendor’s LoC, the SUT partially complies with the rules for server processing of outbound subscription cancellation. The SUT partially complies with requirement IM-001500. Upon receiving the outbound subscription cancellation, the contact’s server does not send a presence stanza of type “unavailable” from all of the contacts online resources to the user.	Minor	See note 2.
Per the vendor’s LoC, the SUT partially complies with the rules for server processing of inbound unsubscribe. The SUT partially complies with requirement IM-001540.	Minor	See note 2.
Per the vendor’s LoC, the SUT does not comply with server generation of inbound presence probe.	Minor	See note 3.
The SUT Unified Presence Server establishes SASL external authentication with the incorrect domain name.	Minor	See note 3.

Table 1. Conditions (continued)

Condition	Operational Impact	Remarks
Open Test Discrepancies (continued)		
The SUT does not comply to the requirements in XMPP Extension XEP-0045 (multi-user chat). The SUT does not host or participate in multi-user chat/chat rooms as required by the reference.	Minor	See note 3.
The SUT Jabber Video Client when calling the Polycom Group series video EI has 1-way audio.	Minor	CLOSED See note 1.
The SUT does not support Local RTS Database (LRDB).	Minor	See note 2.
The SUT does not support Master RTS Database (MRDB).	Minor	See note 2.
During testing for DTR 3, the SUT dropped calls when transferring between two environments with the 99xx series End Instruments (EIs).	Minor	See note 3.
During testing for DTR 6, the SUT 8800 series of IP phones was not tested for IPv4 and IPv6 dual stack functionality. Therefore, the SUT 8800 series IP phones are certified for IPv4 only.	Minor	See note 3.
During testing for DTRs 7 and 9, video calls between the SUT video End Instruments and the Vidyo UCCS drop after 15 minutes. DISA has accepted the vendor's POA&M and has adjudicated this discrepancy as having a minor operational impact.	Minor	See note 9.
During testing for DTRs 7 and 9, video calls that originate from the SUT DX650, SX10, or SX80 to Vidyo UCCS end points have two-way audio and video; however, the Vidyo EI has a severely distorted and fuzzy picture. DISA has adjudicated this discrepancy as having a minor operational impact while this issue is researched to determine the source of the discrepancy.	Minor	See note 9.
During testing for DTR 7, when a video call is originated from the SUT DX-650 to a Polycom Group Series video EI, the call sets up with two-way audio and one-way video. The SUT DX-650 is able to see video and hear audio, but there is no video on the Polycom GS.	Minor	See note 9.
During testing for DTR 7, when a video call is originated from the SUT DX-70 to a Polycom Group Series video EI, the call sets up with one-way audio and two-way video. The SUT DX-70 is able to see video and hear audio, but there is no audio on the Polycom GS.	Minor	See note 9.
During testing for DTR 23, when a call is originated by the Avaya AS5300 Softclient to Cisco 8865, the call sets up with 2-way audio and 2-way video. After a period of time (1-hr or less), the call goes from 2-way audio/video at both endpoints to 1-way audio/video only at the 8865 phone. This is only the case when the Avaya Softclient is the originator of the call. If the Cisco 8865 originates the call, then the call stays up with 2-way audio and 2-way video until the call is hung up. This anomaly did not occur with the 8845,, which has the same software as the 8865.	Minor	See note 10.

NOTES:
1. During the original test, the SUT Jabber Video Client when calling the Polycom Group series video EI had 1-way audio. In the original certification, the Jabber client was certified for audio only as a soft phone and for XMPP IM&P as an XMPP client. The video discrepancy was fixed and the video capability of Jabber client was successfully tested during the DTR 2 test window. See the deployment guide for the configuration change.
2. DISA has adjudicated this discrepancy as minor and stated the intent to change this requirement in the next version of the UCR.
3. DISA has accepted the vendor's POA&M and has adjudicated this discrepancy as minor.
4. During the original test, video calls between SUT H.323 PEIs (C90/EX90/SX20) and other UC video endpoints dropped at approximately 30 minutes. This discrepancy with call drops at 30 minutes was fixed and successfully tested with DTR 1, which included updated VCS software release x8.1.1.
5. DISA has adjudicated this discrepancy as minor and stated the intent to remove this requirement from the UCR and apply it to a DoD STIG.
6. During the original test, the SUT 9951/9971 voice/video SIP ROEIs did not fully support inter-enclave hold feature while video enabled. In the original certification, the 9951/9971 were not covered under this certification. The SUT 29xx and 39xx SBC with IWG was updated from IOS 15.2(4)M5 to 15.2(4)M7 for DTR 8 and this discrepancy was fixed and tested during the DTR 8 test window. Therefore, the 9951/9971 voice/video SIP ROEI is now covered under this certification. In addition, the SUT 29xx and 39xx SBC with IWG software was updated from IOS 15.2(4)M7 to IOS 15.4(3)M2 for DTR 11 and this update also enables the SUT 9951/9971 to support the inter-enclave hold feature for video.
7. This discrepancy applies only to the SUT configured as an ESC. DISA has accepted the vendor's POA&M and has adjudicated this discrepancy as minor. In the interim, the LiteScape EDS Gateway has been successful with posting their LiteScape EDS gateway on the UC APL under tracking number 1412803. LiteScape is certified on the UC APL only with the SUT. The SUT now meets the EDS ESC minimum essential interoperability requirements with the LiteScape EDS Gateway.
8. DISA has adjudicated this discrepancy as minor.
9. DISA has adjudicated this discrepancy as minor while the issue is researched to determine the source of the discrepancy.
10. This TDR was adjudicated by DISA as minor and is being researched to determine the source of the anomaly.

Table 1. Conditions (continued)

LEGEND:			
AEI	AS-SIP End Instrument	LoC	Letter of Compliance
ANSI	American National Standards Institute	MCU	Multipoint Control Unit
APL	Approved Products List	NFAS	Non Facility Associated Signaling
ASAC	Assured Services Admission Control	PEI	Proprietary End Instrument
AS-SIP	Assured Services Session Initiation Protocol	POA&M	Plan of Action and Milestones
CUPS	Cisco Unified Presence Server	RFC	Request for Comments
DISA	Defense Information System Agency	ROEI	ROUTINE Only End Instrument
DN	Directory Number	SASL	Simple Authentication and Security Layer
DTR	Desktop Review	SBC	Session Border Controller
EDS	Enterprise Directory Services	SDP	Session Description Protocol
EI	End Instrument	SIP	Session Initiation Protocol
ESC	Enterprise Session Controller	SUT	System Under Test
ID	identification	STIG	Security Technical Implementation Guide
IM/P	Instant Messaging/Presence	TLS	Transport Layer Security
IOS	Internetwork Operating System	UC	Unified Capabilities
IP	Internet Protocol	UCCS	Unified Capabilities Conference System
IPv4	Internet Protocol version 4	UCR	Unified Capabilities
IPv6	Internet Protocol version 6	VCS	Video Communication Server
IWG	Internetworking Gateway	XMPP	Extensible Messaging and Presence Protocol

Interoperability Status. Table 2 provides the SUT interface interoperability status and Table 3 provides the Capability Requirements (CR) and Functional Requirements (FR) status. Table 4 provides the UC APL product summary.

Table 2. Interface Status

Interface	Threshold CR/FR Requirements (See note.)	Status	Remarks
Network Management Interfaces			
10BaseT (R)	4, 6, 9, 13, 16, 20, 21, 23, 24	Certified	The SUT met the critical CRs and FRs for the IEEE 802.3i interface.
100BaseT (R)	4, 6, 9, 13, 16, 20, 21, 23, 24	Certified	The SUT met the critical CRs and FRs for the IEEE 802.3u interface.
1000BaseT (C)	4, 6, 9, 13, 16, 20, 21, 23, 24	Certified	The SUT met the critical CRs and FRs for the IEEE 802.3ab interface.
Network Interfaces (Line and Trunk)			
10BaseT (R)	1, 5, 6, 7, 8, 10, 11, 13, 14, 15, 17, 18, 19, 20, 21, 22, 23, 24, 25	Certified	The SUT met the critical CRs and FRs for the IEEE 802.3i interface with the SUT PEIs and softphones.
100BaseT (R)	1, 5, 6, 7, 8, 10, 11, 13, 14, 15, 17, 18, 19, 20, 21, 22, 23, 24, 25	Certified	The SUT met the critical CRs and FRs for the IEEE 802.3u interface with the SUT PEIs and softphones.
1000BaseT (R)	1, 5, 6, 7, 8, 10, 11, 13, 14, 15, 17, 18, 19, 20, 21, 22, 23, 24, 25	Certified	The SUT met the critical CRs and FRs for the IEEE 802.3ab interface with the SUT PEIs and softphones.
2-wire analog (R)	1, 8, 15, 17, 19, 20, 21, 22, 23	Certified	The SUT met the critical CRs and FRs for the 2-wire analog interface with the SUT 2-wire secure and non-secure analog instruments.
ISDN BRI (C)	1, 8, 15, 17, 19, 20, 21, 22, 23	Not Tested	The SUT offers this interface; however, it was not tested because it does not support Assured Services and is not required for an ESC.
Legacy Interfaces (External)			
10BaseT (C)	2, 3, 5, 6, 7, 8, 11, 13, 18, 20, 21, 23, 24, 25	Certified	The SUT met the critical CRs/FRs for IEEE 802.3i for the AS-SIP trunk.
100BaseT (C)	2, 3, 5, 6, 7, 8, 11, 13, 18, 20, 21, 23, 24, 25	Certified	The SUT met the critical CRs/FRs for IEEE 802.3u for the AS-SIP trunk.
1000BaseT (C)	2, 3, 5, 6, 7, 8, 11, 13, 18, 20, 21, 23, 24, 25	Certified	The SUT met the critical CRs/FRs for IEEE 802.3ab for the AS-SIP trunk.
ISDN T1 PRI (ANSI T1.619a) (R)	3, 9, 12, 14, 22, 20, 21, 23	Certified	The SUT met the critical CRs/FRs. This interface provides legacy DSN and TELEPORT connectivity.

Table 2. Interface Status (continued)

Interface	Threshold CR/FR Requirements (See note.)	Status	Remarks
Legacy Interfaces (External) (continued)			
ISDN T1 PRI NI-2 (R)	3, 9, 12, 14, 22, 20, 21, 23	Certified	The SUT met the critical CRs/FRs. This interface provides PSTN connectivity.
T1 CCS7 (ANSI T1.619a) (C)	3, 9, 12, 14, 20, 21, 22, 23	Not Tested	The SUT does not support this conditional interface.
T1 CAS (C)	3, 9, 12, 14, 20, 21, 22, 23	Certified	The SUT met threshold CRs/FRs for DTMF.
E1 PRI (ITU-T Q.955.3) (C)	3, 9, 12, 14, 20, 21, 22, 23	Certified	The SUT met the critical CRs/FRs. This interface provides OCONUS MLPP connectivity in ETSI-compliant countries.
E1 PRI (ITU-T Q.931) (C)	3, 9, 12, 14, 20, 21, 22, 23	Certified	The SUT met the critical CRs/FRs. This interface provides OCONUS connectivity in ETSI-compliant countries.

NOTE: The SUT high-level CR and FR ID numbers depicted in the Threshold CRs/FRs column can be cross-referenced in Table 3. These high-level CR/FR requirements refer to a detailed list of requirements provided in Reference (c), Enclosure 3.

LEGEND:

10BaseT	10 Mbps Ethernet	IEEE	Institute of Electrical and Electronics Engineers
100BaseT	100 Mbps Ethernet	ISDN	Integrated Services Digital Network
1000BaseT	1000 Mbps Ethernet	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
ANSI	American National Standards Institute		
AS-SIP	Assured Services Session Initiation Protocol	Mbps	Megabits per second
BRI	Basic Rate Interface	MLPP	Multi-Level Precedence and Preemption
C	Conditional	NI-2	National ISDN Standard 2
CAS	Channel Associated Signaling	OCONUS	Outside the Continental United States
CCS7	Common Channel Signaling Number 7	PEI	Proprietary End Instrument
CR	Capability Requirement	PRI	Primary Rate Interface
DSN	Defense Switched Network	PSTN	Public Switched Telephone Network
DTMF	Dual Tone Multi-Frequency	Q.931	Signaling Standard for ISDN
E1	European Basic Multiplex Rate (2.048 Mbps)	Q.955.3	ISDN Signaling Standard for E1 MLPP
ESC	Enterprise Session Controller	R	Required
ETSI	European Telecommunications Standards Institute	SS7	Signaling System 7
FR	Functional Requirement	T1	Digital Transmission Link Level 1 (1.544 Mbps)
ID	Identification	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1

Table 3. SUT Capability Requirements and Functional Requirements Status

CR/FR ID	UCR Requirement (High-Level) (See note 1.)	UCR 2013 Reference	Status
1	Voice Features and Capabilities (R)	2.2	Partially Met (See note 2.)
2	Assured Services Admission Control (R)	2.3	Met
3	Signaling Protocols (R)	2.4	Met
4	Registration and Authentication (R)	2.5	Met
5	SC and SS Failover and Recovery (R)	2.6	Met
6	Product Interface (R)	2.7	Met
7	Product Physical, Quality, and Environmental Factors (R)	2.8	Met
8	End Instruments (including tones and announcements) (R)	2.9	Partially Met (See note 2.)
9	Session Controller (R)	2.10	Met
10	AS-SIP Gateways (C)	2.11	Met (See note 3.)
11	Enterprise UC Services (R)	2.12	Partially Met (See notes 2 4, and 5.)
12	Call Connection Agent (R)	2.14	Met
13	CCA Interaction with Network Appliances and Functions (R)	2.15	Met
14	Media Gateway (R)	2.16	Met
15	Worldwide Numbering & Dialing Plan (R)	2.18	Met
16	Management of Network Devices (R)	2.19	Partially Met (See note 2.)
17	V.150.1 Modem Relay Secure Phone Support (R)	2.20	Partially Met (See note 2.)
18	Requirements for Supporting AS-SIP Based Ethernet Devices for Voicemail Systems (C)	2.21	Not Tested

Table 3. SUT Capability Requirements and Functional Requirements Status (continued)

CR/FR ID	UCR Requirement (High-Level) (See note 1.)	UCR 2013 Reference	Status
19	Local Attendant Console Features (O)	2.22	Not Tested
20	MSC and SSC (O)	2.23	Not Tested (See note 6.)
21	MSC, SSC, and Dynamic ASAC Requirements in Support of Bandwidth-constrained links (O)	2.24	Not Tested (See note 7.)
22	Other UC Voice (R)	2.25	Partially Met (See note 2.)
23	Information Assurance Requirements (R)	4	Partially Met (See notes 2 and 7.)
24	IPv6 Requirements (R)	5	Partially Met (See note 2.)
25	Assured-Services (AS) Session Initiation Protocol (SIP) (AS-SIP 2013) (R)	AS-SIP	Partially Met (See note 2.)

NOTES:
1. The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Reference (c), Enclosure 3.
2. The SUT met the requirements with the exceptions noted in Table 1. DISA adjudicated these exceptions as minor.
3. During the original test, video calls between SUT H.323 PEIs (C90/EX90/SX20) and other UC video endpoints dropped at approximately 30 minutes. This discrepancy was fixed and successfully tested with DTR 1, which included VCS software release x8.1.1.
4. These requirements apply specifically to an Enterprise Session Controller.
5. During the original test, the SUT Jabber Video Client when calling the Polycom Group series video EI had 1-way audio. In the original certification, the Jabber client was certified for audio only as a soft phone and for XMPP IM&P as an XMPP client. The video discrepancy was fixed and the video capability of Jabber client was successfully tested during the DTR 2 test window. See the deployment guide for the configuration change.
6. This optional requirement applies specifically to a Local Session Controller.
7. Security is tested by DISA-led Information Assurance test teams and the results published in a separate report, Reference (e).

LEGEND:

AS-SIP	Assured Services Session Initiation Protocol	O	Optional
C	Conditional	PEI	Proprietary End Instrument
CCA	Call Connection Agent	R	Required
CR	Capability Requirement	SC	Session Controller
DISA	Defense Information System Agency	SS	Softswitch
DTR	Desktop Review	SUT	System Under Test
FR	Functional Requirement	UC	Unified Capabilities
ID	Identification	UCR	Unified Capabilities Requirements
IPv6	Internet Protocol version 6	VCS	Video Communication Server

Table 4. UC APL Product Summary

Product Identification			
Product Name	Cisco Enterprise Session Controller (ESC) 8		
Software Release	8		
UC Product Type(s)	Enterprise Session Controller (ESC) or Local Session Controller		
Product Description	Enterprise Session Controller for Type 1, 2, and 3 Environments or as a Local Session Controller		
Product Components (See note 1.)	**Component Name (See notes 2 and 3.)**	**Version**	**Remarks**
Unified Communications Manager	Cisco Unified Communications Manager	8.6	See note 4.
Session Management Edition	Cisco Session Management Edition	8.6	See note 4.
Unified Communications Manager	**Cisco Unified Communications Manager**	8.6	See note 4.
Cisco Unity Connection	Cisco Unity Connection	8.6	
Cisco Unified Presence Server	Cisco Unified Presence Server	8.6	
Cisco Webex Meeting Server	Cisco Webex Meeting Server	2.5	See note 5.
E911 management system	RedSky E911 Management System	6.3.1	See note 6.
Interworking Gateway	IWG on 3925 ISR G2, IWG on 3925E ISR G2, **IWG on 3945 ISR G2**, IWG on 3945E ISR G2	IOS 15.4(3)M2	See note 7.
Session Border Controller	SBC on 3925 ISR G2, SBC on 3925E ISR G2, **SBC on 3945 ISR G2**, SBC on 3945E ISR G2	IOS 15.4(3)M2	See note 8.

Table 4. UC APL Product Summary (continued)

Product Components (See note 1.)	Component Name (See notes 2 and 3.)	Version	Remarks
Session Border Controller	**SBC on ISR 4451-X Router**	IOS-XE 3.11	
Session Border Controller	**SBC on ASR 1002**, SBC on ASR 1002-X, SBC on ASR 1004, SBC on ASR 1006	IOS-XE 3.11	
Voice Gateway	2901 ISR G2, 2911 ISR G2, 2921 ISR G2, 2951 ISR G2, 3925 ISR G2, 3925E ISR G2, **3945 ISR G2**, 3945E ISR G2	IOS 15.4(3)M2	See note 9.
Voice Gateway	4321 ISR G3, 4331 ISR G3, 4351 ISR G3, 4431 ISR, G3, **4451 ISR G3**	IOS 15.4(3)M2	See note 10.
Analog Voice Gateway	VG350 Analog Voice Gateway	IOS 15.4(3)M2	See note 11.
Jabber	Cisco Jabber for Windows	9.2	See note 12.
IP Phone (See note 13.)	Unified IP Phone **6901**	9.2.1	
IP Phone (See note 13.)	Unified IP Phone 6911	9.2.1	
IP Phone (See note 13.)	Unified IP Phone 6911	9.2.1	
IP Phone (See note 13.)	Unified IP Phone 6921	9.2.1	
IP Phone (See note 13.)	Unified IP Phone 6941	9.2.1	
IP Phone (See note 13.)	Unified IP Phone **6945**	9.2.1	
IP Phone (See note 13.)	Unified IP Phone **6961**	9.2.1	
IP Phone (See note 13.)	Unified IP Phone 7821	10.1.1.9	
IP Phone (See note 13.)	Unified IP Phone 7841	10.1.1.9	
IP Phone (See note 13.)	Unified IP Phone 7861	10.1.1.9	
IP Phone (See note 13.)	Unified IP Phone 7906G	9.3.1	
IP Phone (See note 13.)	Unified IP Phone **7911G**	9.3.1	
IP Phone (See note 13.)	Unified IP Phone 7931G	9.3.1	
IP Phone (See note 13.)	Unified IP Phone 7941G	9.3.1	
IP Phone (See note 13.)	Unified IP Phone 7941G-GE	9.3.1	
IP Phone (See note 13.)	Unified IP Phone 7942G	9.3.1	
IP Phone (See note 13.)	Unified IP Phone 7945G	9.3.1	
IP Phone (See note 13.)	Unified IP Phone 7961G	9.3.1	
IP Phone (See note 13.)	Unified IP Phone 7961G-GE	9.3.1	
IP Phone (See note 13.)	Unified IP Phone **7962G**	9.3.1	
IP Phone (See note 13.)	Unified IP Phone 7965G	9.3.1	
IP Phone (See note 13.)	Unified IP Phone **7970G**	9.3.1	
IP Phone (See note 13.)	Unified IP Phone 7971G	9.3.1	
IP Phone (See note 13.)	Unified IP Phone 7975G	9.3.1	
IP Phone	Unified IP Phone Expansion Module **7915**	Not Applicable	
IP Phone	Unified IP Phone Expansion Module 7916	Not Applicable	
IP Conference Phone	IP Conference Station 8831	9.3.3.5	
IP Phone (See note 14.)	Unified IP Phone 8811	10.3.1	See note 15.
IP Phone (See note 14.)	Unified IP Phone 8841	10.3.1	See note 15.
IP Phone (See Note 14.)	Unified IP Phone 8845	10.3.2	See note 16.
IP Phone (See note 14.)	Unified IP Phone 8851	10.3.1	See note 15.
Secure Phone	EM-8841-xx and EM-8851-xx	10.3.1	See note 17.
IP Phone (See note 14.)	Unified IP Phone 8851NR	10.3.1	See note 18.
IP Phone (See note 14.)	Unified IP Phone 8861	10.3.1	See note 15.
IP Phone (See Note 14)	Unified IP Phone 8865	10.3.2	See note 16.
IP Phone (See note 14.)	Unified IP Phone 8961	9.4.1	
IP Phone (See note 14.)	Unified IP Phone 9951	9.4.2	See note 7.
IP Phone (See note 14.)	Unified IP Phone 9971	9.4.2	See note 7.

Table 4. UC APL Product Summary (continued)

Product Components (See note 1.)	Component Name (See notes 2 and 3.)	Version	Remarks
IP Phone Expansion Module	Unified IP Color Key Expansion Module	Not Applicable	
Secure Phone	CIS Secure DTD-7965-TSGB	9.3.1	
Secure Phone	CIS Secure DTD-7962-TSG-01	9.3.1	
Secure Phone	CIS Secure DTD-7962-T2	9.3.1	
Secure Phone	API DNT502-02, API DNT-502	9.3.1	See note 19.
Secure Phone	API DNC502-10, API DNC-502	9.3.1	See note 20.
Secure Phone	CIS Secure DTD-7975-T2	9.3.1	See note 21.
Secure Phone	**Telecore 2151**	2AE-00199-0301	
Video Teleconference	TelePresence Video Communication Server (VCS)	X8.1.1	See note 22.
Video Teleconference	TelePresence QuickSet C20	7.2	See note 23.
Video Teleconference	TelePresence Codec C40, TelePresence Codec C60, **TelePresence Codec C90**	7.2	See note 23.
Video Teleconference	TelePresence EX60, TelePresence EX90	7.2	See note 23.
Video Teleconference	TelePresence MX200, TelePresence MX300	7.2	See note 23.
Video Teleconference	**TelePresence SX20 QuickSet,** TelePresence MX300 G2	7.2	See note 23.
Video Teleconference	**Telepresence Quickset SX 10, Telepresemce Codec SX80**	7.2	See note 24.
Video Teleconference	**DX70**, DX80, and **DX650**	10.2.4	See note 25.
Video Teleconference	MX700, MX800	7.2	See note 26.
Video Teleconference	MX200G2, VX-Tactical, VX-Clinical Assistant	7.2	See note 27.
Video Teleconference	**TelePresence 5300 MCU**	4.4(1.68)	
Common Access Card/Single sign-on solution	**OpenAM**	11.0	See note 28.
Common Access Card support for WebEx Meeting Server	Cisco ASA	8.4(3)	
Cisco Wireless phone	CP-7925	1.4.1	See note 29.
Softphone	Cisco IP Communicator	8.6.1.0	See note 30.

NOTES:
1. The detailed component and subcomponent list is provided in Reference (c), Enclosure 3.
2. Components bolded and underlined were tested by JITC. The other components in the family series were not tested but are also certified for joint use. JITC certifies those additional components because they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes.
3. A comprehensive list of supported hardware configurations can be found by selecting the "Cisco Unified Communications on the Cisco Unified Computing System" link at the following URL: www.cisco.com/go/swonly.
4. The SUT was updated from 8.6.1 20011-4 to 8.6(1.20012-14) with DTR 3. The SUT was updated from ES12 (version 8.6(1.20012-14)) to ES14 (version 8.6(1.20014-8)) with DTR 20.
5. The SUT Cisco Webex Meeting Server (CWMS) was updated from Release 2.0 to 2.5 with DTR 4. This CWMS update includes support for preset conferencing and security revisions that included Single Sign On capability via Security Assertion Markup Language verison 2.0, which were successfully tested with DTR 4. In addition, DTR 4 documents the removal of the Cisco MeetingPlace Server from the SUT.
6. The SUT is certified with any RedSky E911 Management system or other E911 Management system listed on the UC APL and certified with the Cisco UCM. E911 management is not required for an LSC.
7. During the original test, the SUT 9951/9971 voice/video SIP ROEIs did not fully support inter-enclave hold feature while video enabled. In the original certification, the 9951/9971 were not covered under this certification. The SUT 29xx and 39xx SBC with IWG was updated from IOS 15.2(4)M5 to 15.2(4)M7 for DTR 8 and this discrepancy was fixed and tested during the DTR 8 test window. Therefore, the 9951/9971 voice/video SIP ROEI is now covered under this certification. In addition, the SUT 29xx and 39xx SBC with IWG software was updated from IOS 15.2(4)M7 to IOS 15.4(3)M2 for DTR 11 and this update also enables the SUT 9951/9971 to support the inter-enclave hold feature for video.
8. The SUT 39xx series SBC was updated from IOS 15.2(4)M5 to IOS 15.4(3)M2 with DTR 11.
9. The SUT 29xx/39xx series Voice Gateway was updated from IOS 15.2(4)M5 to IOS 15.4(3)M2 with DTR 12.
10. The 4321 ISR G3, 4331 ISR G3, 4351 ISR G3, 4431 ISR, G3, and 4451 ISR G3 with release IOS 15.4(3)M2 were included with DTR 21. The 4451 ISR G3 was tested during the V&V for DTR 21. The other components in the family series were not tested but are also certified for joint use. JITC certifies those additional components because they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes
11. The SUT VG350 Voice Gateway was updated from IOS 15.2(4)M5 to IOS 15.4(3)M2 with DTR 13.

Table 4. UC APL Product Summary (continued)

NOTES (continued):

12. During the original test, the SUT Jabber Video Client when calling the Polycom Group series video EI had 1-way audio. In the original certification, the Jabber client was certified for audio only as a soft phone and for XMPP IM&P as an XMPP client. The video discrepancy was fixed and the video capability of Jabber client was successfully tested during the DTR 2 test window. See the deployment guide for the configuration change.
13. Although this IP phone supports SIP and SCCP protocols, only the SCCP protocol was tested and is covered under this certification.
14. This IP phone was tested and is certified for use with the SIP protocol.
15. The 8800 series of IP phones was included with DTR 6 as Routine Only End Instruments (ROEI). DTR 6 testing was conducted on the 8811 and 8841 phones, but the IPv4 and IPv6 dual stack functionality was not tested due to infrastructure limitations. DISA has accepted the vendor's POA&M and has adjudicated this discrepancy as minor. Therefore, the SUT 8800 series IP ROEI phones are certified for IPv4 only. JITC analysis also determined the 8851 and 8861 phones were functionally identical for purposes of interoperability to the tested 8800 series phones, and therefore they are included as certified products with DTR 6.
16. The 8845 and 8865 VVoIP phones were tested and included with DTR 23. There was one new discrepancy during this test. When a video call is originated by the Avaya AS5300 Session Controller (SC) Softclient to Cisco 8865, the call sets up with 2-way audio and 2-way video. After a period of time (1-hr or less), the call goes from 2-way audio/video at both endpoints to 1-way audio/video only at the 8865 phone. This only occurs when the Avaya AS5300 Softclient is the originator of the call. If the Cisco 8865 originates the call, then the call stays up with 2-way audio and 2-way video until the call is hung up. This anomaly did not occur with the 8845, which has the same software as the 8865.. This TDR was adjudicated by DISA as minor and is being researched to determine the source of the anomaly.
17. The EM-8841-xx and EM-8851-xx phones are modified Cisco 8841 and 8851 IP phones that have enhanced security when placed on-hook, in call hold mode, or when muted. The EM-8841-xx and EM-8851-xx phones are Telecommunications Security Group (TSG) accredited to Committee on National Security Systems (CNSS) 5001 defined requirements and TEMPEST Level 1 certified. The JITC analysis determined EM-8841-xx and EM-8851-xx phones are functionally identical to the certified Cisco 8841 and 8851 IP phones and therefore they are certified without testing in DTR18.
18. The 8851NR IP phone was included with DTR 22. JITC analysis determined the 8851NR IP phone is identical in regards to interoperability and IA posture to the certified Cisco 8851 phone except the 8851NR IP phone does not support Bluetooth.
19. The API DNT502-02 and API DNT-502 TEMPEST 7962G IP phones were included with DTR 16.
20. The API DNC502-10 and API DNC-502 TEMPEST 7962G IP phones were included with DTR 17.
21. The CIS Secure DTD-7975-T2 phone is a modified Cisco 7975G IP phone that has enhanced security when placed on-hook, in call hold mode, or when muted. The DTD-7975-T2 phone is Telecommunications Security Group (TSG) accredited to Committee on National Security Systems (CNSS) 5001 defined requirements and TEMPEST Level 1 certified. The JITC analysis determined the DTD-7975-T2 phone is functionally identical to the certified 7975G IP phone and therefore they is certified without testing in DTR24.
22. The VCS release was updated from x7.2.2 to x8.1.1 with DTR 1.
23. This codec was updated from Release TC 7.1.1 to 7.2 with DTR 9.
24. The SX10 and SX80 were included with DTR 9. Noted discrepancies are discussed in paragraph 4 of DTR 9.
25. The DX70, DX80, and DX650 were included with DTR 7. JITC analysis determined the DX80 video phone was functionally identical to the DX70 and DX650 tested phones and, therefore, the DX80 is also certified as part of DTR 7 without testing. Noted discrepancies are discussed in paragraph 4 of DTR 7.
26. The MX700 and MX800 were included with DTR 9 based upon JITC similarity analysis to the SX80.
27. The MX200G2, VX-Tactical, and VX-Clinical Assistant were included with DTR 9 based upon JITC similarity analysis to the SX20.
28. The OpenAM version was updated from 9.5.5 to 11.0 with DTR 5.
29. The Cisco CP-7925 Wireless phone with release 1.4.1 was tested and certified with the Cisco UCM Release 8.6.1 (20010-5) as a Local Session Controller under Unified Capabilities Certification Office (UCCO) Tracking Number 1108301 and has been added to this certification in DTR 25 without additional testing based on JITC analysis.
30. The Cisco IP Communicator with release 8.6.1.0 was tested and certified with the Cisco UCM Release 8.6.1 (20010-5) as a Local Session Controller under Unified Capabilities Certification Office (UCCO) Tracking Number 1108301 and has been added to this certification in DTR 25 without additional testing based on JITC analysis.

LEGEND:

API	Advanced Programs Inc
APL	Approved Products List
DISA	Defense Information System Agency
DTR	Desktop Review
G2	Generation 2
IM/P	Instant Messaging/Presence
IOS	Internetwork Operating System
IP	Internet Protocol
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
ISR	Integrated Services Router
IWG	Interworking Gateway
JITC	Joint Interoperability Test Command
MCU	Multipoint Conference Unit
POA&M	Plan of Action and Milestones
ROEI	ROUTINE Only End Instrument
SBC	Session Border Controller
SC	Session Controller
SCCP	Skinny Call Control Protocol
SIP	Session Initiation Protocol
UC	Unified Capabilities
UCM	Unified Communications Manager
VCS	Video Communication Server
VVoIP	Voice and Video over Internet Protocol
XMPP	Extensible Messaging and Presence Protocol

Test Details. The original certification, documented in Reference (c), is based on interoperability testing, DISA adjudication of open test discrepancy reports (TDRs), review of the vendor's Letters of Compliance (LoC), and DISA Certifying Authority (CA)

Recommendation for inclusion on the UC APL. The initial testing was conducted under UCCO Tracking Number 1108301 from 11 July through 5 August 2011 on the SUT as an LSC. Additional testing of the LSC under UCCO Tracking Number 1108301 was conducted for Desktop Reviews and documented in extensions to the original certification. Testing was conducted from 7 April through 9 May 2014 on the Cisco UCM as an ESC. The data from the LSC test is included in this certification. The test procedures derived from the UCR Reference (b) using test procedures derived from Reference (d) were used to validate the deltas between a Local Session Controller (LSC) and an ESC. Review of the vendor's LoC was completed on 7 April 2014. DISA adjudication of outstanding TDRs was completed on 10 June 2014. Information Assurance (IA) testing was conducted by DISA-led Information Assurance test teams and the results are published in a separate report, Reference (e).

The extension of this certification is based upon DTR 23. This DTR was requested to include the 8845 and 8865 VVoIP phones. JITC determined IA and interoperability Verification & Validation (V&V) testing was required.

JITC conducted interoperability testing from 14 through 25 September 2015. There was one new interoperability Test Discrepancy Report (TDR) documented during this V&V test. When a video call is originated by the Avaya AS5300 Session Controller (SC) Softclient to Cisco 8865, the call sets up with 2-way audio and 2-way video. After a period of time (1-hr or less), the call goes from 2-way audio/video at both endpoints to 1-way audio/video only at the 8865 phone. This only occurs when the Avaya AS5300 Softclient is the originator of the call. If the Cisco 8865 originates the call, then the call stays up with 2-way audio and 2-way video until the call is hung up. This anomaly also did not occur with the Cisco 8845, which has the same software as the 8865.

This TDR was adjudicated by DISA as minor and is being researched to determine the source of the anomaly. There were no interoperability discrepancies closed as a result of this V&V test. DISA adjudication was completed on 29 September 2015. Information Assurance testing was also conducted by DISA-led Information Assurance test teams and the results are published in a separate report, Reference (e). Therefore, JITC approves this DTR.

Additional Information. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Sensitive but Unclassified IP Data (formerly known as NIPRNet) e-mail. Interoperability status information is available via the JITC System Tracking Program (STP). STP is accessible by .mil/.gov users at https://stp.fhu.disa.mil/. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at https://jit.fhu.disa.mil/. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly from the Unified Capabilities Connection Office (UCCO), e-mail: disa.meade.ns.list.unified-capabilities-certification-office@mail.mil. All associated information is available on the DISA UCCO website located at http://www.disa.mil/Services/Network-Services/UCCO.

Point of Contact (POC). The JITC point of contact is Mr. Joseph Schulte, commercial telephone (520) 538-5100, DSN telephone 879-5100, FAX DSN 879-4347; e-mail address joseph.t.schulte.civ@mail.mil; mailing address Joint Interoperability Test Command, ATTN: JTE (Mr. Joseph Schulte) P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The UCCO tracking number for the SUT is 1331201.

FOR THE COMMANDER:

Enclosure a/s

for RIC HARRISON
Chief
Networks/Communications and UC Portfolio

Distribution (electronic mail):
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UCCO

ADDITIONAL REFERENCES

(c) Joint Interoperability Test Command, Memo, JTE, "Joint Interoperability Certification of the Cisco Enterprise Session Controller (ESC) 8," 13 June 2014
(d) Joint Interoperability Test Command, "Enterprise Session Controller (ESC) Test Procedures for Unified Capabilities Requirements (UCR) 2013," Draft
(e) Joint Interoperability Test Command, "Information Assurance (IA) Findings Summary For Cisco ESC 8 (Tracking Number 1331201)," Draft